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PRECAUTIONS

PLEASE READ CAREFULLY BEFORE PROCEEDING

* Please keep this manual in a safe place for future reference.

🗥 WARNING

Always follow the basic precautions listed below to avoid the possibility of serious injury or even death from electrical shock, short-circuiting, damages, fire or other hazards. These precautions include, but are not limited to, the following:

Power supply/Power cord

- Only use the voltage specified as correct for the device. The required voltage is printed on the name plate of the device.
- Use only the specified AC power adaptor (PA-20) or an equivalent recommended by PfLE PR0).
- Do not place the power cord near heat sources such as heaters or radiators, and do not excessively bend or otherwise damage the cord, place heavy objects on it, or place it in a position where anyone could walk on, trip over, or roll anything over it.

Do not open

 Do not open the device or attempt to disassemble the internal parts or modify them in any way. The device contains no user-serviceable parts. If it should appear to be malfunctioning, discontinue use immediately and have it inspected by qualified PME PR0 service personnel.

Water warning

- Do not expose the device to rain, use it near water or in damp or wet conditions, or place containers on it containing liquids which might spill into any openings.
- · Never insert or remove an electric plug with wet hands.

If you notice any abnormality

- If the power cord or plug becomes hayed or damaged, or if there is a sudden loss of sound during use of the device, or if any unsual smells or snoke should appear to be caused by al, immediately turn off the power switch, disconnect the electric plug from the outile, and have the device inspected by qualified PLFE observice personnel.
- If this device or the AC power adaptor should be dropped or damaged, immediately turn off the power switch, disconnect the electric plug from the outlet, and have the device inspected by qualified PNE PRO service personnel.

riangle caution

Always follow the basic precautions listed below to avoid the possibility of physical injury to you or others, or damage to the device or other property. These precautions include, but are not limited to, the following:

Power supply/Power cord

- Remove the electric plug from the outlet when the device is not to be used for extended periods of time, or during electrical storms.
- When removing the electric plug from the device or an outlet, always hold the plug itself and not the cord. Pulling by the cord can damage it.
- To avoid generating unwanted noise, make sure there is 50cm or more between the AC power adaptor and the device.
- · Do not cover or wrap the AC power adaptor with a cloth or blanket.

Location

- · Before moving the device, remove all connected cables.
- When setting up the device, make sure that the AC outlet you are using is easily
 accessible. If some trouble or malfunction occurs, immediately turn off the
 power switch and disconnect the plug from the outlet.
- Avoid setting all equalizer controls and faders to their maximum. Depending on the condition of the connected devices, doing so may cause leedback and may damage the speakers.
- Do not expose the device to excessive dust or vibrations, or extreme cold or heat (such as in direct sunlight, near a heater, or in a car during the day) to prevent the possibility of panel disfiguration or damage to the internal components.
- Do not place the device in an unstable position where it might accidentally tall over.

 Do not use the device in the vicinity of a TV, radio, stereo equipment, mobile phone, or other electric devices. Doing so may result in noise, both in the device itself and in the TV or radio next to it.

Connections

 Before connecting the device to other devices, turn off the power for all devices. Before turning the power on or off for all devices, set all volume levels to minimum.

Handling caution

- When turning on the AC power in your audio system, always turn on the power amplifier LAST, to avoid speaker damage. When turning the power off, the power amplifier should be turned off FIRST for the same reason.
- Do not insert your fingers or hands in any gaps or openings on the device.
- Avoid inserting or dropping foreign objects (paper, plastic, metal, etc.) into any gaps or openings on the device III this trappens, turn off the power immediately and unplug the power cord from the AC outlet. Then have the device inspected by qualified PEEPR0 service personnel.
- Do not use the device or headphones for a long period of time at a high or uncomfortable volume level, since this can cause permanent hearing loss. If you experience any hearing loss or ringing in the ears, consult a physician.
- Do not rest your weight on the device or place heavy objects on it, and avoid use excessive force on the buttons, switches or connectors.

XLR-type connectors are wired as follows (IEC60268 standard): pin 1: ground, pin 2: hot (+), and pin 3: cold (-), Insert TRS phone jacks are wired as follows: sleeve: ground, tip: send, and ring: return.

Always turn the power off when the device is not in use.

Even when the power switch is in the "STANDBY" position, electricity is still flowing to the device at the minimum level. When you are not using the device for a long time, make sure you unplug the power cord from the wall AC outlet.

The performance of components with moving contacts, such as switches, volume controls, and connectors, deteriorates over time. Consult qualified PEE PRO service personnel about replacing detective components.

The PMX mixer may heat up by as much as 15 to 20°C while the power is on. This is normal. Please note that the panel temperature may exceed 50°C in ambient temperatures higher than 30°C, and use caution to prevent burns.

* Illustrations herein are for explanatory purposes only, and may not match actual appearance during operation.

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Specifications and descriptions in this owner's manual are for information purposes only PI(EPR0, reserves the right to change or modify products or specifications at any time without prior notice. Since specifications, equipment or options may not be the same in every locale, please check with your PI(EPR0 dealer.

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Introduction

Thank you for your purchase of the PYLEPRO PMX1205 mixing console. The PMX1205 feature input channels suitable for a wide range of usage environments. And the PMX1205 includes high-quality built-in digital effects that can provide some very serious sound. The mixer combines ease of operation with support for multiple usage environments.

Please read through this manual carefully before beginning use, so that you will be able to take full advantage of this mixer's superlative features and enjoy trouble-free operation for years to come.

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Accessories

- Owner's Manual
- AC power adaptor (PA-20)*
 - * May not be included depending on your particular area. Please check with your PYLEPRO dealer.

Before Turning on the Mixer

Be sure that the mixer's power switch is in the STANDBY position.

Use only the PA-20 adaptor included with this mixer. Use of a different adaptor may result in equipment damage, overheating, or fire.

Connect the power adaptor to the AC ADAPTOR IN connector (③) on the rear of the mixer, and then turn the fastening ring clockwise (②) to secure the connection.



3 Plug the power adaptor into a standard household power outlet.

 Be sure to unplug the adaptor from the outlet when not using the mixer, or when there are lightning storms in the area.

 To avoid generating unwanted noise, make sure there is 50 cm or more between the power adaptor and the mixer.

Turning the Power On

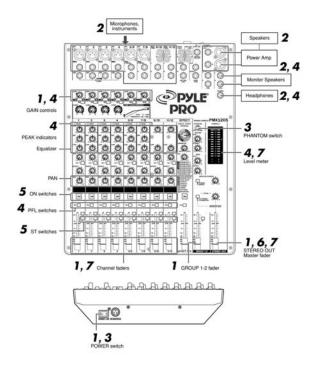
Press the mixer's power switch to the ON position. When you are ready to turn the power off, press the power switch to the STANDBY position.



Note that trace current continues to flow while the switch is in the STANDBY position. If you do not plan to use the mixer again for a long while, please be sure to unplug the adaptor from the wall outlet.

Getting Sound to the Speakers

We begin by connecting a pair of speakers and generating some stereo output. Note that operations and procedures will vary somewhat according to the input devices you are using.





Quick Guide

Be sure that your mixer is turned off and that all level* controls are turned all the way down.

* STEREO OUT Master Fader, Channel Fader, GROUP 1-2 Fader, Gain Control, etc.

NOTE

Set the equalizer and the pan controls to their V positions.

Turn off any other external devices, then connect microphones, instruments, and speakers.

NOTE

- * For information on connecting external devices see the Connection Example on page 11.
- * Connect electric guitars and basses through an intermediary device such as a direct box, preamp, or amp simulator. Connecting these instruments directly to the MG mixer may result in degraded sound and noise.
- 3 To avoid damage to your speakers. power up the devices in the following order: Peripheral devices → PMX mixer → power amps (or powered speakers). Reverse this order when turning power off.

NOTE

If you are using microphones that require phantom power, turn the PMX mixer's phantom power switch on before turning on the power to the power amp or powered speakers. See page 15 for more detail.

Adjust the channel GAIN controls so that the corresponding peak indicators flash briefly on the highest peak levels.

NOTE

To use the LEVEL meter to get an accurate reading of the incoming signal level, turn the channel PFL switch on. Adjust the GAIN controls so that the LEVEL meter indication occasionally rises above the "T" (0) level. Note that the PHONES jack outputs the pre-fader signal from all channels on which the PFL switch is ON so that those signals can be monitored via the headphones.

- 5 Turn on the ON and ST switches for each channel you are using.
 - Set the STEREO OUT Master fader to the "0" position.
 - Set the Channel faders to create the desired initial balance, then adjust the overall volume using the STEREO **OUT Master fader.**

NOTE

- * To use the LEVEL meter to view the level being applied to the STEREO L/R buses, set the PFL REO(.)
- * If the PEAK indicator lights frequently, lower the Channel faders a little to avoid distortion.

Making the Most of Your Mixer

You've got yourself a mixer and now you're ready to use it.

Just plug everything in, twiddle the controls, and away you go ... right?

Well, if you've done this before you won't have any problems, but if this is the first time you've ever used a mixer you might want to read through this little tutorial and pick up a few basics that will help you get better performance and make better mixes.

Balanced, Unbalanced—What's the Difference?

In a word: "roise," The whole point of balanced lines is noise rejection, and it's something they're very goot at. Any length of wire will act as an antenna to pick up the random electromagnetic radiation we're constantly surrounded by: radio and TV signals as well as spurious electromagnetic noise generated by power lines, motors, electric appliances, computer monitors, and a variety of other sources. The longer the wire, the more noise it is likely to pick up. That's why balanced lines are the best choice for long cable runs. If your "studio" is basically confined to your desktop and all connections are no more than a meter or two in length, then unbalanced lines are fine—unless you're surrounded by extremely high levels of electromagnetic noise. Another place balanced lines are almost always used is in microphone cables. The reason for this is that the output signal from most microphones is very small, so even a tiny amount of noise will be relatively large, and will be amolifier.

Balanced noise cancellation

To summarize

	-	4			s
A 0	Hot (+)	hal) hh	$ \land \land \land$	n
*	Cold (-)	XX [Noise-free	
Phase inversion	Ground	00	Phase Inversion &	signal	L
Source	L L Ca	ble	Noise cancelled Receiving device		
				1.50	

Microphones:	Use balanced lines.
Short line-level runs:	Unbalanced lines are fine if you're in a relatively noise-free environment.
Long line-level runs:	The ambient electromagnetic noise level will be the ultimate deciding factor, but balanced is best.

Signal Levels and the Decibel

Let's take a look at one of the most commonly used units in audio: the decide) (dB). If the smallest sound that can be heard by the human ear is given an arbitrary value of 1, then the loudest sound that can be heard is approximately 1,000,000 (one million) times louder. That's too many digits to deal with for pactical calculations, and so the more appropriate direcibel' (dB) unit was created for sound-related measurements. In this system the difference between the softest and loudest sounds that can be heard is 120 dB. This is a non-linear scale, and a difference of 3 dB actually results in a doubling or halving of the loudness.

You might encounter a number of different varieties of the dB: dBu, dBV, dBM and others, but the dBu is the basic decibel unit. In the case of dBu, '0 dBu' is specified as a signal level of 0.775 volts. For example, if a microphone's output level is ~40 dBu (0.00775 V), then to raise that level to dBu (0.775 V) in the mixer's preamp stage requires that the signal be amplified by 100 times.



A mixer may be required to handle signals at a wide range of levels, and it is necessary match input and output levels as closely as possible. In most cases the "nominal" level for a mixer's input and outputs is marked on the panel or listed in the owner's manual.

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Making the Most of Your Mixer

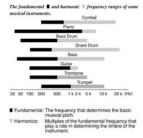
To EQ or Not to EQ

In general: less is better. There are many situations in which you'll need to cut certain frequency ranges, but use boost sparingly, and with caution. Proper use of EQ can eliminate interference between instruments in a mix and give the overall sound better definition. Bad EQ—and most commonly bad boost—just sounds terrible.

Cut for a Cleaner Mix

For example: cymbals have a lot of energy in the mid and low frequency ranges that you don't really perceive as musical sound, but which can interfere with the clarity of other instruments in these ranges. You can basically turn the low EQ on cymbal channels all the way down without changing the way they sound in the mix. You'll hear the difference, however, in the way the mix sounds more "spacious," and instruments in the lower ranges will have better definition. Surprisingly enough, piano also has an incredibly powerful low end that can benefit from a bit of low-frequency roll-off to let other instruments notably drums and bass—do their jobs more effectively. Naturally you won't want to do this if the piano is playing solo.

The reverse applies to kick drums and bass guitars: you can often roll off the high end to create more space in the mix without compromising the character of the instruments. You'll have to use your ears, though, because each instrument is different and sometimes you'll want the 'snap' of a bass guitar, for example, to come through.



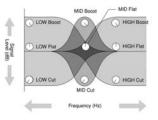
Some Frequency Facts

The lowest and highest frequencies than can be heard by the human ear are generally considered to be around 20 Hz and 20,000 Hz, respectively. Average conversation occurs in the range from about 300 Hz to about 30,000 Hz. The frequency of a standard pitchford, used to nume guitars and other instruments is 440 Hz (this corresponds to the "A" key on a piano tuned to concert pitch). Double this frequency to 880 Hz and you have a pitch one octave higher (i.e. "Ad" on the piano keyboard), in the same way you can have the frequency to 280 Hz and you have a pitch one octave higher (i.e."

Boost with Caution

If you're trying to create special or unusual effects, go ahead and boost away as much as you like. But if you're just trying to achieve a goodsounding mix, boost only in very small increments. A tiny boost in the midrange can give vocals more presence, or a touch of high boost can give certain instruments more "air." Listen, and if things don't sound clear and clean try using cut to remove frequencies that are cluttering up the mix rather than trying to boost the mix into clarity.

One of the biggest problems with too much boost is that it adds gain to the signal, increasing noise and potentially overloading the subsequent circuitry.



Making the Most of Your Mixer

Ambience

Your mixes can be further refined by adding ambience effects such as reverb or delay. The PMX's internal effects can be used to add reverb or delay to individual channels in the same way as external effects processors. (Refer to page 16).

Reverb and Delay Time

The optimum reverb time for a piece of music will depend on the music's tempo and density, but as a general rule longer reverb times are good for ballads, while shorter reverb times are more suited to up-tempo tunes. Delay times can be adjusted to create a wide variety of "grooves". When adding delay to a vocal, for example, try setting the delay time to dotted eighth notes corresponding to the tune's tempo.

Reverb Tone

Different reverb programs will have different "reverb tone" due to differences in the reverb time of the high or low frequencies. Too much reverb, particularly in the high frequencies, can result in unnatural sound and interfere with the high frequencies in other parts of the mix. It's always a good idea to choose a reverb program that gives you the depth you want without detracting from the clarity of the mix.

Reverb Level

It's amazing how quickly your ears can lose perspective and fool you into believing that a totally washed-out mix sounds perfectly fine. To avoid falling into this trap start with reverb level all the way down, then gradually bring the reverb into the mix until you can just hear the difference. Any more than this normally becomes a "special effect."

The Modulation Effects: Phasing, Chorus, and Flanging

All of these effects work on basically the same principle: a portion of the audio signal is "timeshifted" and then mixed back with the direct signal. The amount of time shift is controlled, or "modulated", by an LFO (Low-frequency Oscillator). For phasing effects the shift is very small. The phase difference between the modulated and direct signals causes cancellation at some frequencies and reinforces the signal at others and this causes the shimmering sound we hear.

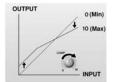
For chorus and flanging the signal is delayed by several milliseconds, with the delay time modulated by an LFO, and recombined with the direct signal. In addition to the phasing effect described above, the delay modulation causes a perceived pitch shift which, when mixed with the direct signal, results in a harmonically rich swirling or swishing sound.

The difference between chorus and flanging effects is primarily in the amount of delay time and feedback used-flanging uses longer delay times than chorus, whereas chorus generally uses a more complex delay structure. Chorus is most often used to thicken the sound of an instrument, while flanging is usually used as an outright "special effect" to produce otherworldly sonic swoops.

Compression

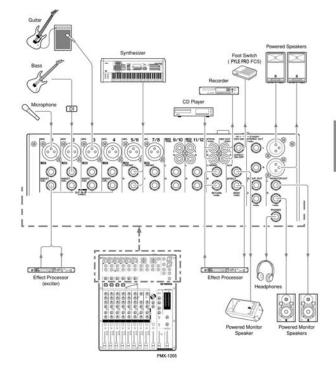
One form of compression known as "limiting" can. when properly used, produce a smooth, unified sound with no excessive peaks or distortion. A common example of the use of compression is to "tame" a vocal that has a wide dynamic range in order to tighten up the mix. With the right amount of compression you'll be able to clearly hear whispered passages while passionate shouts are still well balanced in the mix. Compression can also be valuable on bass guitar. Too much compression can be a cause of feedback, however, so use it sparingly.

Most compressors require several critical parameters to be set properly to achieve the desired sound. The PMX compressor makes achieving great sound much easier: all you need to do is set a single "compression" control and all of the pertinent parameters are automatically adjusted for you.



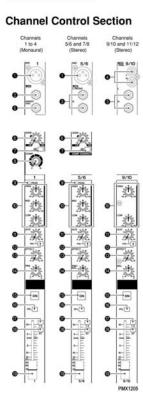
Reference

Setup



Front & Rear Panels

The following applies to the PMX1205.



MIC Input Jacks (CHs 1 to 4, 5/6, 7/8) These are balanced XLR-type microphone input jacks (1:Ground: 2:Hot: 3:Cold).

A LINE Input Jacks (CHs 1 to 4)

These are balanced TRS phone-jack line inputs (T:Hot; R:Cold: S:Ground). You can connect either balanced or unbalanced phone plues to these jacks.

CLINE Input Jacks (CHs 5/6 to 11/12) These are unbalanced phone-iack stereo line inputs.

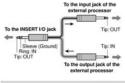
These are unbalanced stereo RCA pin jacks.

NOTE Where an input channel provides both a MIC input jack and a LINE input jack, or a LINE input jack and an RCA pin jack, you can use either jack but not both at the same time. Please connect to only one jack on each channel.

INSERT Jacks (CHs 1 to 4)

Each of these jacks provides an insert point between the equalizer and fader of the corresponding input channel (CHs 1 to 4). The INSERT jacks can be used to independently connect devices such as graphic equalizers, compressors, or noise filters into the corresponding channels. These are TRS (tip, ring, sleeve) phone jacks that carry both the send and return signal (tip = send/out; ring = return/in; sleeve = ground).

Patching external devices via an INSERT jack requires a special insert cable such as illustrated below (insert cable sold separately).



The signal output from the INSERT jacks is reverse-phased. This should not be a problem when connecting to an effect unit, but please be aware of the possibility of phase conflict when connecting to other types of device.

G GAIN Control

Adjusts the input signal level.

To achieve the best balance between S/N ratio and dynamic range, adjust the level so that the PEAK indicator () lights only occasionally and briefly on the highest input transients. The -60 to -16 scale is the MIC input adjustment range. The -34 to +10 scale is the LINE input adjustment range.



1/80 Switch (High Pass Filter)

This switch toggles the HPF on or off. To turn the HPF on, press the switch in (....). The HPF cuts frequencies below 80 Hz (the HPF does not apply to the line inputs of stereo input channels ().

COMP Control

Adjusts the amount of compression applied to the channel. As the knob is turned to the right the compression ratio increases while the output gain is automatically adjusted accordingly. The result is smoother, more even dynamics because louder signals are attenuated while the overall level is boosted.

NOTE Avoid setting the compression too high, as the the higher average output level that results may lead to feedback.

PEAK Indicator

The peak level of the post-EQ signal is detected, and the PEAK indicator lights red when the level reaches 3 dB below clipping. For XLR-equipped stereo input channels (5/6 and 7/8), both the post-EQ and post-mic-amp peak levels are detected, and the indicator lights red if either of these levels reaches 3 dB below clipping.

Equalizer (HIGH, MID, and LOW)

This three-band equalizer adjusts the channel's high, mid, and low frequency bands. Channels 9/10 and 11/12 have two bands: high and low. Setting the knob to the V position produces a flat response in the corresponding band. Turning the knob to the right boosts the corresponding frequency band, while turning to the left attenuates the band. The following table shows the EQ type, frequency, and maximum cut/boost for each of the three hands.

Band	Туре	Frequency	Maximum Cut/Boost
HIGH	Shelving	10 kHz	
MID	Peaking	2.5 kHz	±15 dB
LOW	Shelving	100 Hz	1

AUX (AUX1) Control

Adjusts the level of the signal sent from the channel to the AUX (AUX1) bus. The knob should generally be set close to the V position.

On stereo channels, the signals from the L (odd) and R (even) channels are mixed and sent to the AUX (AUX1) bus.

NOTE To send the signal to the buses set the ON switch to on (.m.).

AUX PRE Switch

Selects whether the pre-fader or the post-fader signal is fed to the AUX (AUX1) bus. If the switch is on (.....), the mixer sends the pre-fader signal (the signal immediately prior to the Channel fader (1) to the AUX (AUX1) bus, so that AUX (AUX1) output is not affected by the fader. If the switch is off (I) the mixer sends the post-fader signal to the AUX (AUX1) bus.

EFFECT (AUX2) Controls

Adjusts the level of the signal sent from the channel to the EFFECT (AUX2) bus. Note that the signal level sent to the bus is also affected by the Channel fader. On stereo channels (5/6, 7/8, 9/10, or 11/12), the signals from the L (odd) and R (even) channels are mixed and then sent to the EFFECT (AUX2) bus,

PAN Control (1 to 4) PAN/BAL Control (5/6 and 7/8) BAL Control (9/10 and 11/12)

The PAN control determines the stereo positioning of the channel signal on the Group 1 and 2 buses or on the Stereo L. and R buses.

The BAL control knob sets the balance between left and right channels. Signals input to the L input (odd channel) go to the Group 1 bus or to the Stereo L bus; signals input to the R input (even channel) go to the Group 2 bus or the Stereo R bus.



NOTE On channels where this knob provides both PAN and BAL control (channels 5/6 and 7/8), the knob operates as a PAN control when input is received via the MIC jack or L (MONO) input only, and as a BAL control when input is received via both L and R inputs.

ON Switch

Turn this switch on to send the signal to the buses. The switch lights orange when on.

PFL (Pre-Fader Listen) Switch

This switch lets you monitor the channel's pre-fader signal. Press the switch in (.....) so that it lights to turn it on. When the switch is on the channel pre-fader signal is output to the PHONES and MONITOR OUT (iacks for monitoring.

1-2 Switch

This switch assigns the channel's signal to the Group I and 2 buses.

NOTE To send the signal to the Group buses engage the ON switch (.m.),

ST Switch

This switch assigns the channel's signal to the Stereo L and R hoses

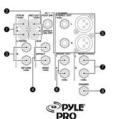
NOTE To send the signal to the Stereo buses engage the ON switch (.....)

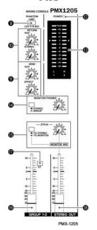
Channel Fader

Adjusts the level of the channel signal. Use these faders to adjust the balance between the various channels.

NOT Set the fader sliders for unused channels all the way down to minimize noise.

Master Control Section





* impedance balanced

Since the hot and cold terminals of impedance balanced output jacks have the same impedance, these output jacks are less affected by induced noise.

2TR IN Jacks

These RCA pin jacks can be used to input a stereo sound source. Use these jacks when you want to connect a CD player directly to the mixer.

NOTE You can adjust the signal level using the 2TR IN control in the Master Control section.

REC OUT (L, R) Jacks

These RCA pin jacks can be connected to an external recorder such as an MD recorder in order to record the same signal that is being output via the STEREO OUT jacks.

NOTE The mixer's STEREO OUT Master Fader has no affect on the signal output via these jacks. Be sure to make appropriate level adjustments at the recording device

RETURN L (MONO), R Jacks

These are unbalanced phone-iack type line inputs. The signal received by these jacks is sent to the STEREO L/R buses and the AUX (AUX1) bus. These jacks are typically used to receive the signal returned from an external effect device (reverb, delay, etc.).

NOTE These jacks can also be used as an auxiliary stereo input. If you connect to the L (MONO) jack only, the mixer will recognize the signal as monaural and will send the identical signal to both the L and R jacks.

SEND Jacks

AUX (AUX1)

This is an impedance balanced* phone-jack type output. This jack outputs the signals from AUX (AUX1) bus. You can use this jack, for example, to connect to an effect unit, cue box, or other monitoring system.

EFFECT (AUX2)

This is an impedance balanced* phone-jack type output that outputs the signal from the EFFECT (AUX2) bus. You can use this jack, for example, to connect to an external effect unit.

STEREO OUT (L, R) Jacks

These jacks deliver the mixer's stereo output. You can use these jacks, for example, to connect to the power amplifier driving your main speakers. You can also connect these jacks to a recording device when you wish to record mixer's stereo output while using the STEREO OUT Master fader for level control.

XLR jacks

XLR-type balanced output jacks.

LINE jacks

TRS phone-jack type balanced outputs.

G GROUP OUT (1, 2) Jacks

These impedance-balanced* phone jacks output the Group 1/2 signals. Use these jacks to connect to the inputs of a multi-track recorder, external mixer, or other such device,

MONITOR OUT Jacks

Connect these stereo phone-jack outputs to your monitor system.

NOTE The signal output by these jacks is determined by the MONITOR switch, the 2TR IN switch, and the PFL switches on the input channels.



Front & Rear Panels

PHONES Jack

Connect a pair of headphones to this stereo phone jack. The PHONES jack outputs the same signal as the MONITOR OUT jacks.

PHANTOM +48 V Switch

This switch toggles phantom power on and off. When the switch is on the mixer supplies +48V phantom power to all channels that have XLR mic input jacks (CHs 1-4, 5/6, 7/8).

Turn this switch on when using one or more phantom-powered condenser microphones.

NOTE When this switch is on the mixer supplies DC +48 V power to pins 2 and 3 of all XLR-type MIC INPUT jacks.



 Be sure to leave this switch off (IL) if you do not need phantom power.

- When tuning the switch on (.....), be sure that only condenser mics are connected to the XLR input jacks (CHs: 1 to 7/8). Devices other than condenser mics may be damaged if connected to the phantom power supply. Note, however, that the switch may be left on when connecting to balanced dynamic microphones.
- · To avoid damage to speakers, be sure to turn off amplifiers (or powered speakers) before turning this switch on or off. We also recommend that you turn all output controls (STE-REO OUT Master Fader, GROUP 1-2 Fader, etc.) to their minimum settings before operating the switch to avoid the risk of loud noises that could cause hearing loss or device damage.

RETURN

· AUX (AUX1) Control

Adjusts the level at which the L/R signal received at the RETURN jacks (L (MONO) and R) is sent to the AUX (AUX2) bus,

STEREO Control

Adjusts the level at which the signal received at the RETURN jacks (L (MONO) and R) is sent to the STEREO L/R buses.

NOTE If you supply a signal to the RETURN L (MONO) jack only, the mixer sends the same signal to both the L and R Stereo buses.

Master SEND

Master AUX (AUX1) Control

Adjusts the signal level sent to the AUX (AUX1) SEND jack.

Master EFFECT (AUX2) Control

Adjusts the level of the signal sent to the EFFECT (AUX2) here

NOTE If you are using the PMX-1205, the Master EFFECT control does not affect the level of the signal sent from the EFFECT bus to the internal digital effect processor.

POWER Indicator

This indicator lights when the mixer's power is ON.

D Level Meter

This LED meter displays the level of the signal selected by the MONITOR switch (), 2TR IN switch () and PFL switch. The "0" segment corresponds to the nominal output level. The PEAK segment lights red when the output reaches the clipping level.

MONITOR/PHONES

MONITOR Switch

If this switch is set to GROUP (____), the Group 1/2 bus signals are sent to the MONITOR OUT jacks, the PHONES STEREO L/R bus signals are sent to these jacks and the level meter

MONITOR Control

Controls the level of the signal output to the PHONES jack and the MONITOR OUT jacks.

2TR IN

2TR IN Switch

If this switch is set to TO MONITOR (....), the signals input via the 2TR IN jacks are sent to the MONITOR OUT jacks, the PHONES jack, and the level meter. If it is set to TO STE-REO (_, the signals are sent to the STEREO L/R buses.

2TR IN control

Adjusts the level of the signal sent from the 2TR IN jacks to the STEREO L/R buses.

The following illustration shows how the switch settings correspond to the signal selection.

	Switche	15	Signals output via the
PFL	MONITOR/ PHONES	2TR IN	MONITOR/PHONES jacks
ON	<u> </u>	-	PFL
OFF STEREO	TO STEREO	STEREO (+ 2TR IN)	
		STEREO + 2TR IN [MONITOR MIX] *	
GROUP	GROUP	TO STEREO	GROUP
-		TO MONITOR	GROUP (+ 2TR IN)

* MONITOR MIX : When overdubbing, you can adjust the levels of the monitor playback signal and the signal being recorded separately.

MONITOR MIX Signal Flow



NOTE If the input channel PFL switch is on (.....), then only the PFL output from that channel is sent to the C-R OUT jacks, PHONES jacks, and level meter.

GROUP 1-2 Fader

Adjusts the signal level sent to the GROUP OUT jacks.

ST Switch

If this switch is on (.....), the signals are sent to the STEREO L/R buses via the GROUP 1-2 fader (. The Group 1 signal goes to Stereo L and the Group 2 signal goes to Stereo R.

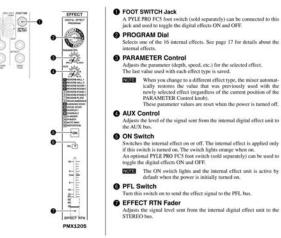
STEREO OUT Master Fader

Adjusts the signal level sent to the STEREO OUT jacks.

Reference ///

Front & Rear Panels

DIGITAL EFFECT



Rear Input/Output Section



POWER Switch

Use this switch to turn the mixer's power ON or to STANDBY mode.



Note that a small current continues to flow while the switch is in the STANDBY position. If you do not plan to use the mixer for a while, be sure to unplug the AC adaptor from the wall outlet.

AC ADAPTOR IN Connector

Connect the supplied PA-20 power adaptor to this connector (see page 5).



Use only the PA-20 adaptor included with this mixer. Use of a different adaptor may result in fire or electric shock.

Reference

Front & Rear Panels

Digital Effect Program List

No	Program	Parameter	Description
1	REVERB HALL 1	REVERB TIME	Break de later a la service de la service de la
2	REVERB HALL 2	REVERB TIME	 Reverb simulating a large space such as a concert hall.
3	REVERB ROOM 1	REVERB TIME	Reverb simulating the acoustics of a small space (room).
4	REVERB ROOM 2	REVERB TIME	Hevero simulating the accustors of a small space (room).
5	REVERB STAGE 1	REVERB TIME	Reverb simulating a large stage.
6	REVERB STAGE 2	REVERB TIME	Heverb simulating a large stage.
7	REVERB PLATE	REVERB TIME	Simulation of a metal-plate reverb unit, producing a more hard-edged sound.
8	DRUM AMBIENCE	REVERB TIME	A short reverb that is ideal for use with kick drum.
9	KARAOKE ECHO	DELAY TIME	Echo designed for karaoke (sing-along) applications.
10	VOCAL ECHO	DELAY TIME	Echo suitable for vocals.
11	CHORUS 1	LFO Frequency	Creates a thick sound by modulating the delay time.
12	CHORUS 2	LFO Frequency	 The PARAMETER control adjusts the frequency of the LFO* that modulates the delay time.
13	FLANGER	LFO Frequency	A sweeping pitched effect. The PARAMETER control adjusts the frequency of the LFO* that modulates the delay time.
14	PHASER	LFO Frequency	Phase modulation produces a cyclical phasing effect. The PARAMETER control adjusts the frequency of the LFO* that modulates the delay time.
15	HAW OTUA	LFO Frequency	A wah-wah effect with cyclical filter modulation. The PARAMETER control adjusts the frequency of the LFO* that modulates the delay time.
16	DISTORTION	DRIVE	Adds a sharp-edged distortion to the sound.

* "LFO" stands for Low Frequency Oscillator. An LFO is normally used to modulate another signal, determining the modulation speed and waveform shape.

Jack List

Input and Output Jacks	Input and Output Jacks Polarities Con	
MIC INPUT, STEREO OUT	Pin 1: Ground Pin 2: Hot (+) Pin 3: Cold (-)	
LINE INPUT(CH1 to 4) GROUP OUT, STEREO OUT, MONITOR OUT, AUX (AUX1), EFFECT (AUX2)*	Tip: Hot (+) Ring: Cold (-) Sleeve: Ground	Bing
INSERT	Tip: Output Ring: Input Sleeve: Ground	جىلىر
PHONES	Tip: L Ring: R Sleeve: Ground	Sleeve Tip
RETURN LINE INPUT (CHSi6 to 11/12) Sileeve: Ground		Sieve Tip

* These jacks will also accept connection to monaural phone plugs. If you use monaural plugs, the connection will be unbalanced.

Troubleshooting

Power doesn't come on.	 Is the supplied power adaptor properly plugged into an appropriate AC wall outlet? Is the supplied power adaptor properly plugged into the mixer? 			
No sound.	Are microphones, external devices, and speakers connected correctly? Are the ON switch and ST switch of the channels you are using turned ON? Are the channel GAIN controls, Channel fader, STEREO OUT master fader and GROUP 1-2 fader set to appropriate levels? Are the MONITOR switch and 2TR IN switch set property? Are your speaker cables connected properly, car are they shorted? If the above checks do not identify the problem, call PYLE PR0 for service. (Refer to the page 71 for a list of service centers.)			
Sound is faint, dis- torted, or noisy.	Are the channel GAIN controls, Channel fader, STEREO OUT master fader and GROUP 1-2 fader set to appropriate levels? Are two different instruments connected to the XLR-type and phone jacks, or to the phone and RCA pin jacks on one channel? Please connect to only one of these jacks on each channel. Is the input signal from the connected device set to an appropriate level? Are you applying the effects at an appropriate level? Are microphones connected to the MIC input jacks on channels 1 to 7/8? If you are using condenser microphones, is the PHANTOM +48 V switch turmed ON?			
 No effect is applied. (If you are using PMX-1205) 	Check that the EFFECT control on each channel is correctly adjusted. Be sure that the internal effect unit's ON switch is turned ON. Be sure that the EFFECT PARAMETER control and EFFECT RTN fader are correctly adjusted.			
 I want spoken words to be heard more clearly. 	Be sure that the /60 switches are ON. Adjust the equalizers (HIGH, MID and LOW) on each channel.			
 I want to output a monitor signal through speakers. 	Connect a powered speaker to the AUX (AUX1) jack* and turn the PRE switch on each channel ON. Then adjust the output signal by using the AUX (AUX1) controls on each channel and the Master SEND control.			

Specifications

Electrical Specifications

			MIN	TYP	MAX	UNIT
Frequency Response		GAIN: min (CHs 1-7/8)				
	GROUP OUT	20 Hz-20 kHz				
	EFFECT/AUX (AUX1, 2*) SEND	Nominal output level @1 kHz Input: CHs 1 to 11/12, RETURN, 2TR IN	-3.0	0.0	1.0	dB
	MONITOR OUT, REC OUT					
Total Harmonic Distortion (THD + N)	STEREO OUT	+14 dBu @ 20 Hz-20 kHz, Input GAIN Control at minimum			0.1	%
Hum & Noise Hum & Noise are measured	CH INPUT 1-4 MIC	EIN (Equivalent Input Noise): Rs = 150 Ω, GAIN: maximum		1	-128	
Hum & Noise are measured with a 6 dB/octave filter @	STEREO OUT	STEREO OUT, GROUP Master fader at nominal			1000	1
12.7 kHz; equivalent to a 20	GROUP OUT	level and all channels' ST and 1-2 switches off.			-88	
kHz filter with infinite dB/octave attenuation.		Master EFFECT/AUX (AUX1, 2) control at nominal level and all CH EFFECT/AUX (AUX1, 2) controls at minimum.			-81	dBu
	STEREO OUT	STEREO OUT, GROUP Master fader and one CH			-64	1
	GROUP OUT	fader at nominal level.			-04	
	STEREO OUT	Residual Output Noise			-98	1
Crosstalk (1 kHz)	Adjacent Input				-70	dB
		STEREO L/R, CHs 1-4, PAN: panned hard left or right			-70	UB
Maximum voltage gain (1 kHz)	Rs = 150 Ω	MIC to CH INSERT OUT		60		
All taders and controls are	INPUT GAIN: maximum	MIC to STEREO OUT	-	84		1
maximum when measured.		MIC to GROUP OUT		04		
PAN/BAL: panned hard left		MIC to GROUP to ST		94		1
or hard right		MIC to REC OUT		62.2		1
		MIC to MONITOR OUT, ST TO MONITOR	-	94		1
		MIC to PHONES OUT		83		1
		MIC to AUX (AUX1*) SEND PRE		76		1
		MIC to AUX (AUX1*) SEND POST, EFFECT (AUX2*) SEND		86		dB
		CH 5/6, 7/8 LINE to STEREO OUT	- 58		1 00	
		CH 5/6, 7/8 LINE to GROUP OUT				
		CH 5/6, 7/8 AUX (AUX1*) SEND PRE		47		1
	CH 5/6, 7/8 LINE to AUX (AUX1*) SEND PO EFFECT (AUX2*) SEND	CH 5/6, 7/8 LINE to AUX (AUX1*) SEND POST, EFFECT (AUX2*) SEND		57		
		CH 9/10, 11/12 to STEREO OUT		34		1
		CH 9/10, 11/12 to GROUP OUT	1	34		
	Rs = 150 Ω	RETURN to STEREO OUT		16		1
		RETURN to EFFECT (AUX2*) SEND		9		1
	Rs = 600 Ω	2TR IN to STEREO OUT		27.8		1
Phantom Voltage	MIC	no load	1	48		V

General Specifications

Input HPF	CHs 1-7/8, 80 Hz, 12 dB/oct
Input equalization CHs 1-7/8 ±15 dB maximum Turn over/roll-off frequency	HIGH: 10 kHz (shelving) MID: 2.5 kHz (paaking) LOW: 100 Hz (shelving)
of shelving: 3 dB blow maxi- mum variable level. CH 9/10-11/12	HIGH: 10 kHz (shelving) LOW: 100 Hz (shelving)
PEAK Indicator	Red LED turns on when post EQ signal (either post MIC HA or post EQ signal for CHs 5/6, 7/8) reaches –3 dB below clipping (+17 dBu).
Internal Digital Effect*	16 PROGRAM, PARAMETER control Foot Switch (Digital Effect On/Off)
LED Level Meter Pre MONITOR Level	2x12 points LED meter (PEAK, +10, +6, +3, 0, -3, -6, -10, -15, -20, -25, -30 dB) PEAK lights if the signal level reaches 3 dB below the clipping level.
Power Supply Adaptor PA-20	AC 35 VCT, 0.94 A, Cable Length = 3.6 m
Power Consumption	30 W
Dimensions (W x H x D)	346.2 mm x 86.1 mm x 436.6 mm
Net Weight	3.2 kg

All faders are nominal if not specified.

Output impedance of signal generator: 150 ohms

Input Specifications

Input Connectors	Gain	Input Impedance	Appropriate Impedance	Sensitivity *	Nominal Level	Max. before Clip- ping	Connector Specifica- tions
CH INPUT MIC (CHs 1-4)	-60 dB	3kΩ	50-600Ω Mics	-80 dBu (0.078 mV)	-60 dBu (0.775 mV)	-40 dBu (7.75 mV)	XLR-3-31 type (bal- anced [1 = GND, 2 = HOT, 3 = COLD])
	-16 dB			-36 dBu (12.3 mV)	-16 dBu (123 mV)	+4 dBu (1.23 V)	
CH INPUT LINE (CHs 1-4)	-34 dB	10kΩ	600Ω Lines	-54 dBu (1.55 mV)	-34 dBu (15.5 mV)	-14 dBu (155 mV)	TRS phone jack (balanced [Tip = HOT, Ring = COLD, Sleeve = GND])
	+10 dB			-10 dBu (245 mV)	+10 dBu (2.45 V)	+30 dBu (24.5 V)	
ST CH MIC INPUT (CHs 5/6, 7/8)	-60 dB	3kΩ	50-600Ω Mics	-80 dBu (0.078 mV)	-60 dBu (0.775 mV)	-40 dBu (7.75 mV)	XLR-3-31 type (bal- anced [1 = GND, 2 = HOT, 3 = COLD])
	-16 dB			-36 dBu (12.3 mV)	-16 dBu (123 mV)	-6 dBu (389 mV)	
ST CH LINE INPUT (CHs 5/6, 7/8)	-34 dB	10kΩ	600Ω Lines	-54 dBu (1.55 mV)	-34 dBu (15.5 mV)	-14 dBu (155 mV)	Phone jack (unbalanced)
	+10 dB			-10 dBu (245 mV)	+10 dBu (2.45 V)	+30 dBu (24.5 V)	
ST CH INPUT (CHs 9/10, 11/12)	(-)	10kΩ	600Ω Lines	-30 dBu (24.5 mV)	-10 dBu (245 mV)	+10 dBu (2.45 V)	Phone jack (unbal- anced) RCA pin jack
CH INSERT IN (CHs 1-4)	,	10kΩ	600Ω Lines	–20 dBu (77.5 mV)	0 dBu (0.775 V)	+20 dBu (7.75 V)	TRS phone Jack (unbalanced [Tip = Out, Ring = In, Sleeve = GND])
RETURN (L, R)	-	10kΩ	600Ω Lines	-12 dBu (195 mV)	+4 dBu (1.23 V)	+24 dBu (12.3 V)	Phone jack (unbalanced)
2TR IN (L, R)	-	10kΩ	600Ω Lines	-26 dBV (50.1 mV)	-10dBV (0.316V)	+10dBV (3.16 V)	RCA pin jack

Where 0 dBu = 0.775 Vrms and 0 dBV= 1 Vrms

* Sensitivity : The lowest level that will produce an output of +4 dB (1.23 V), or the nominal output level when the unit is set to the maximum level. (All faders and level controls are at their maximum position.)

Output Specifications

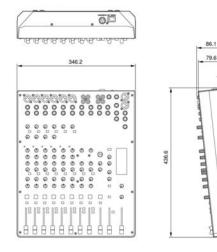
Output Connectors	Output Impedance	Appropriate Impedance	Nominal Level	Max. before clipping	Connector Specifications	
STEREO OUT (L, R)	75Ω	600Ω Lines	+4dBu (1.23 V)	+24 dBu (12.3 V)	XLR-3-32 type (balanced [1 = GND, 2 = HOT, 3 = COLD]) Phone Jack (balanced [Tip = HOT, Ring = COLD, Sleeve = GND])	
3ROUP OUT (1, 2) 1500		10kQ Lines	+4dBu (1.23 V)	+20 dBu (7.75 V)	Phone jack (impedance balanced [Tip = HOT, Ring = COLD, Sleeve = GND])	
EFFECT/AUX (AUX1, 2") SEND 150D		10kΩ Lines	+4dBu (1.23 V)	+20 dBu (7.75 V)	Phone jack (impedance balanced [Tip = HOT, Ring = COLD, Sleeve = GND])	
CH INSERT OUT (CHs 1-4)	75Ω	10kΩ Lines	0 dBu (0.775 V)	+20 dBu (7.75 V)	Phone jack (unbalanced [Tip = Out, Ring = In, Sleeve = GND])	
REC OUT (L, R)	600Ω	10kΩ Lines	-10 dBV (0.316 V)	+10 dBV (3.16 V)	V (3.16 V) RCA pin jack	
MONITOR OUT (L, R)	1500	10kΩ Lines	+4 dBu (1.23 V)	+20 dBu (7.75 V)	Phone jack (impedance balanced [Tip = HOT, Ring = COLD, Sleeve = GND])	
PHONES OUT	1000	400 Phones	3 mW	75 mW	Stereo phone jack	

Where 0 dBu = 0.775 Vrms and 0 dBV= 1 Vrms



Specifications

Dimensional Diagrams



Unit: mm

133.9

2



Block Diagram and Level Diagram

